

User's Manual



SIP IP Phone

▶ VIP-256T



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CE mark Warning

This is a class B device. In a domestic environment, this product may cause radio interference, in which case the user may be required to take adequate measures.

Energy Saving Note of the Device

This power required device does not support Stand by mode operation. For energy saving, please remove the DC-plug or push the hardware Power Switch to OFF position to disconnect the device from the power circuit. Without removing the DC-plug or switching off the device, the device will still consume power from the power circuit. In view of saving the energy and reducing the unnecessary power consumption, it is strongly suggested to switch off or remove the DC-plug for the device if this device is not intended to be active.

WEEE Warning



To avoid the potential effects on the environment and human health as a result of the presence of hazardous substances in electrical and electronic equipment, end users of electrical and electronic equipment should understand the meaning of the crossed-out wheeled bin symbol. Do not dispose of WEEE as unsorted municipal waste; they should be collected separately.

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Revision

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Table of Contents

Chapter 1 Introduction.....	6
Overview	6
Product Features	10
Specifications	11
Package Content.....	13
Physical Details.....	13
Front View and Keypad function	13
Rear View.....	14
Phone Screen Features	15
Chapter 2 Preparations & Installation	17
Physical Installation.....	17
Administration Interface.....	19
Web configuration access.....	19
Chapter 3 Network Service Configurations.....	20
Configuring and monitoring your IP Phone from web browser.....	20
Manipulation of IP Phone via web browser	20
Chapter 4 VoIP IP Phone Status	21
Status	21
Basic.....	21
DHCP	23
Syslog.....	23
Chapter 5 SIP Account Setting	25
SIP Account.....	25
SIP setting	25
NAT setting	26
Line settings.....	27
Basic.....	27
Audio Configuration	28
Supplementary Services Subscription.....	29
Advanced	29
Chapter 6 Network Setting	31
Basic	31
Internet Port (WAN).....	32
PC Port(LAN)	34
Network Address Server Settings (DHCP)	35
MAC Address Clone	36

VPN	37
DMZ	38
QoS	39
Chapter 7 Phone Configurations.....	40
Performance	40
Volume	40
Regional	41
Call Forward	42
Miscellaneous	42
Multi-Functional Key	43
Dial Plan.....	45
Phonebook	49
Phonebook.....	49
Black List.....	50
Call Log	51
Chapter 8 VoIP IP Phone Administration	52
Management	52
Time/Date.....	52
Password Reset	54
Web Access	55
System Log Setting	55
Factory Defaults.....	57
Update Firmware	57
Security	57
Provision	59
SNMP	62
TR 069.....	63
Appendix A Frequently Asked Questions List.....	64

Chapter 1

Introduction

Overview

Cost-effective, High-performance VoIP Phone

To build high-performance VoIP communications at a low cost, PLANET now introduces the latest member of its IP Phone family, the VIP-256T 2-Line Business IP Phone. The VIP-256T makes it simple for the enterprise featuring voice and data system or expanding voice system to new locations. It helps the company to save money on long distance calls; for example, the remote workers can dial in through a Unified VoIP Communication System just like an extension call but no long distance charge would occur. The VIP-256T also allows call to be transferred to anyone at any location within the voice system, which enables the enterprise to communicate more effectively and is helpful to streamline business processes.



Standard Compliance

Compliant with the Session Initiation Protocol 2.0 (RFC 3261), the VIP-256T is able to broadly interoperate with equipment provided by VoIP infrastructure providers, thus enabling them to provide their customers with better multi-media exchange services.

Compliant with standard SIP RFC 3261



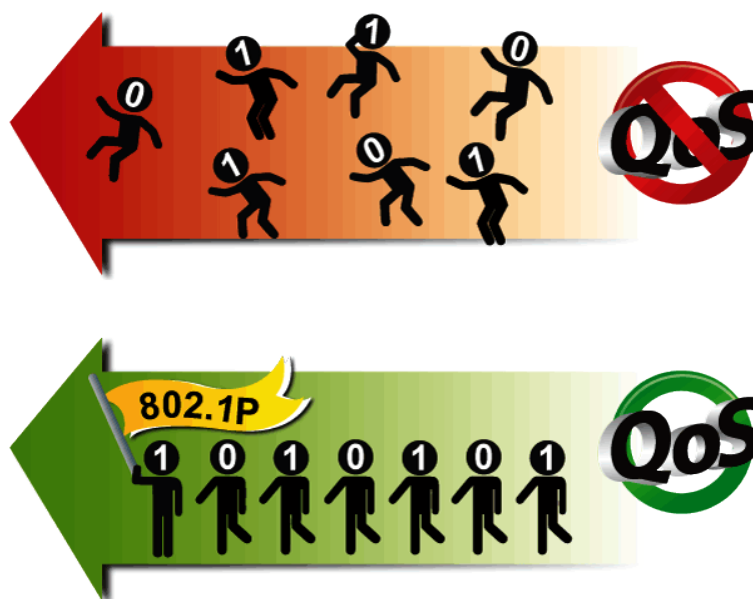
Enhanced, Full-Featured Business IP Phone

The VIP-256T is a full-featured enhanced business IP Phone that addresses the communication needs of the enterprises. It provides 2 voice lines and dual 10/100Mbps Ethernet. Furthermore, the VIP-256T delivers user-friendly design containing a 128x32 Graphic LCD with white backlight, 2 line keys and 4 soft keys, and 10 multi-functional keys with dual-color LED. The VIP-256T supports all kinds of SIP based phone features including Call Waiting, Auto Answer, Music on Hold, Caller ID and Call Waiting ID, 3-way Conferencing, Call Hold, Call Forwarding, Black List, Hotline, DTMF Relay, In-Band, Out-of-Band (RFC 2833) and SIP INFO, among others. Besides for office use, the VIP-256T is also the ideal solution for VoIP service offered by Internet Telephony Service Provider (ITSP).



Secure, High-Quality VoIP Communication

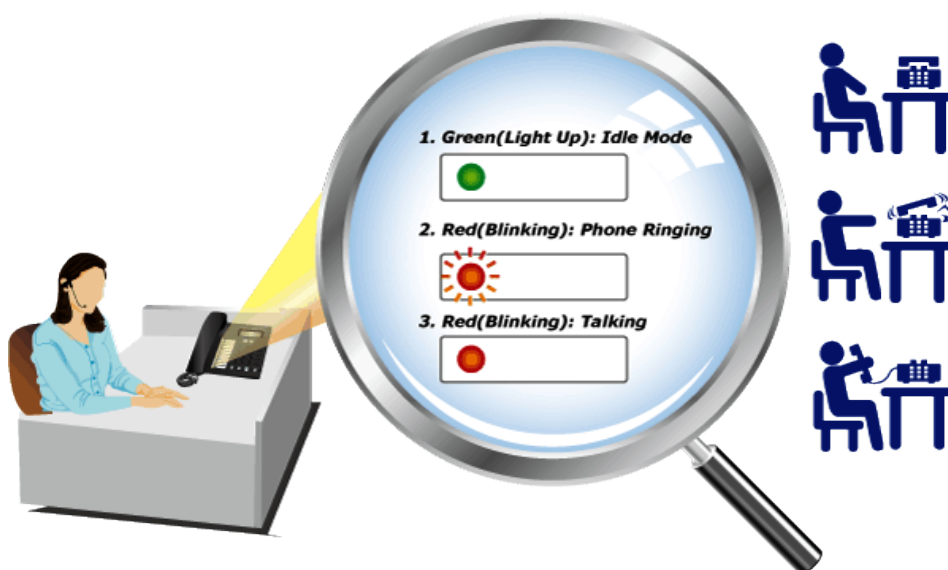
The VIP-256T supports SIP v2 for easy integration with general voice over IP system. It can also effortlessly deliver secured toll voice quality by utilizing cutting-edge 802.1p QoS (Quality of Service), 802.1Q VLAN tagging, and IP ToS technology.



Professional Application

The VIP-256T supports Busy Lamp Field (BLF) function that, via the lights on the phone, enables users to easily identify the status of other phones which are connected to the same IP PBX, such as busy, idle, ringing, etc. The connected IP PBX must also support BLF feature. The BLF function is helpful for a receptionist on the front desk to route all incoming calls smoothly.

BLF (Busy Lamp Field)



Product Features

➤ Highlights

- 2-Line enterprise-class IP phone
- Connects directly to an Internet telephone service provider or to an IP PBX
- Dual switched Ethernet ports, speakerphone, caller ID, call hold, conferencing, and more
- Easy installation and highly secure remote provisioning, as well as menu-based and web-based configuration

➤ Telephony Features

- Two Voice Lines
- Pixel-based Display: 128 x 32 monochrome graphical liquid crystal display (LCD)
- Caller ID and Call Waiting ID
- 3-way Conferencing
- SMS Functions
- Busy Lamp Field (BLF)
- Do Not Disturb (DND)
- Full-Duplex Speakerphone
- Call Transfer: Blind Transfer and Attended Transfer
- Call Mute, Redial, Speed Dial, Pick Up, Call Park, Dial Plan, Call Hold, Call Forwarding

➤ Management

- SIP v2 (RFC 3261, 3262, 3263, 3264)
- STUN (RFC 3489)
- Automated provisioning and upgrade via HTTPS, HTTP, TFTP
- Message Waiting Indicator (RFC 3842)
- IEEE 802.1Q VLAN / 802.1p and IP ToS
- TR069 / SNMP v2

Specifications

Product	VIP-256T SIP IP Phone
Hardware	
Lines (Direct Numbers)	2-Line Enterprise-class IP phone
Display	128x32 Graphic LCD with White Backlight
Feature Keys	2 Line Keys and 4 Soft Keys 12 Dialing Buttons (0~9, *, #) 9 Fixed Function Buttons 10 Multi-functional Key with Dual-Color LED
Physical Interfaces	Two 10/100BASE-T RJ-45 Ethernet Ports (IEEE 802.3) Handset: RJ-9 Connector Built-in Speakerphone and Microphone
Protocols and Standard	
Data Networking	MAC Address (IEEE 802.3) IPv4 (RFC 791) Address Resolution Protocol (ARP) DNS: A Record (RFC 1706), SRV Record (RFC 2782) Dynamic Host Configuration Protocol (DHCP) Client (RFC 2131) Internet Control Message Protocol (ICMP) (RFC 792) TCP (RFC 793) User Datagram Protocol UDP (RFC 768) Real Time Protocol RTP (RFC 1889, 1890) Real Time Control Protocol (RTCP) (RFC 1889) Differentiated Services (DiffServ) (RFC 2475) Type of Service (ToS) (RFC 791, 1349) VLAN Tagging 802.1p / 802.1Q: Layer 2 Quality of Service (QoS) Simple Network Time Protocol (SNTP) (RFC 2030) Backward Compatible with RFC 2543 Session Timer (RFC 4028) SDP (RFC 2327) NAPTR for SIP URI Lookup (RFC 2915)
Voice Gateway	SIP Version 2 (RFC 3261, 3262, 3263, 3264) SIP supported in NAT Networks [including STUN (RFC 3489)] Message Waiting Indicator (RFC 3842) Voice Algorithms: - G.711 (A-law and μ -law) - G.726 (16/24/32/40 kbps) - G.722 - G.723 Dual-Tone Multi-Frequency (DTMF), In-Band and Out-of-Band (RFC 2833) (SIP INFO) Voice Activity Detection (VAD) with Silence Suppression Adaptive Jitter Buffer Management Comfort Noise Generation Echo Cancellation Message
Provisioning, Administration, and Maintenance	Integrated web server provides web-based administration and configuration Telephone Keypad Configuration via Display Menu/Navigation Automated Provisioning and Upgrade via HTTPS, HTTP, TFTP User Authentication for Configuration Pages Local and Remote Syslog (RFC 3164) SNTP Time Synchronization Multi User Level SNMP v2 TR069
Features	
Telephony Features	Two Voice Lines Call Waiting

	Auto Answer Music on Hold Caller ID and Call Waiting ID 3-way Conferencing Call Hold, Call Forwarding, Call Mute Call Transfer: Blind Transfer and Attended Transfer Call Log: Redial List, Answered Calls and Missed Calls Volume Adjustment: Handset/Headset, Speaker and Ringer Delayed Hotline Redial, Speed Dial Busy Lamp Field (BLF) Pick Up, Call Park, Dial Plan Black List Message Waiting Indicator (MWI) Do Not Disturb (DND) Full-Duplex Speakerphone Customized Ring Tone SMS (100 records) Call History (100 records) - Most Recently Missed Calls - Most Recently Received Calls - Most Recently Dialed Numbers Phone Book (100 records) Speed Dial (10 records)
Environment	
Power Requirements	5V DC, 1A
Operating Temperature	0 ~ 50 Degree C
Operating Humidity	10 ~ 90% (non-condensing)
Weight	720g
Dimensions (W x D x H)	191 x 205 x 75 mm
Emission	CE, FCC
Connectors	Two 10/100Mbps Ethernet, RJ-45 RJ-9 handset connector DC power jack DND Switch

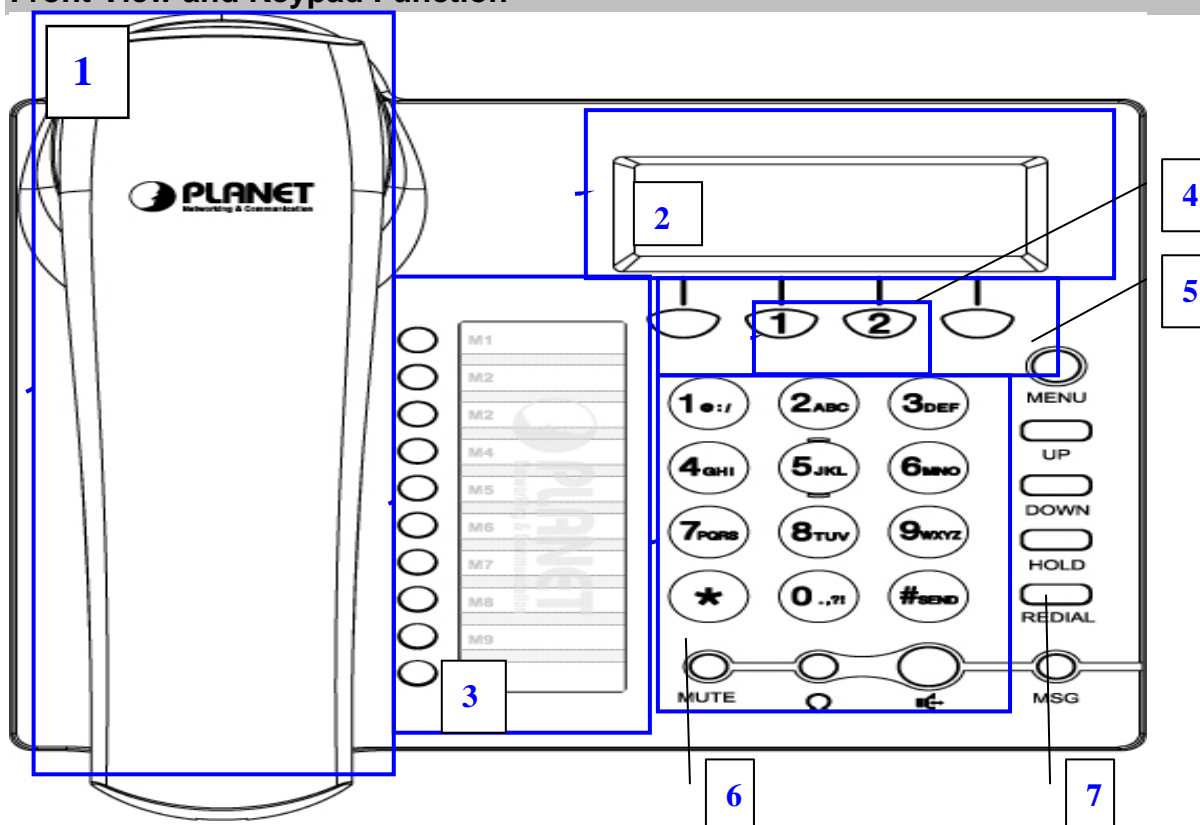
Package Content

- SIP IP Phone unit
- Power Adapter
- Quick Installation Guide
- CD-ROM containing the on-line manual.
- RJ-45 cable x1

Physical Details

The following figure illustrates the front/rear panel of IP Phone.

Front View and Keypad Function

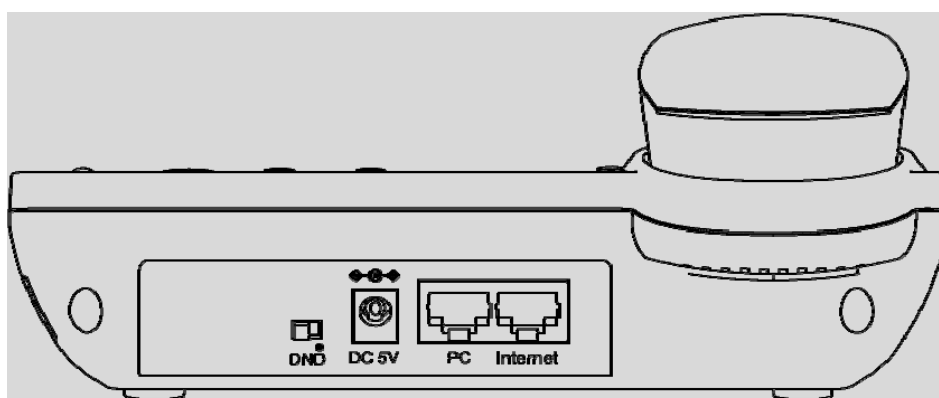
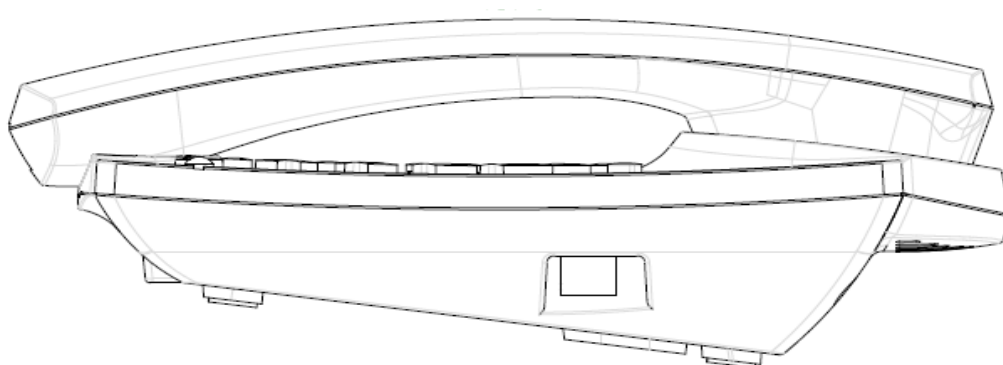


Keypad Descriptions

1	Handset Top Cradle	For the placement of handset (Receiver end)
	Hook Switch	For hang-up and hang-off of handset
	Cradle Latch	To prevent the handset from dropping when it is wall-mounted.
	Handset Bottom Cradle	For the placement of handset (Transmitter end)
	Handset Cord Port	RJ-11 jack on the left side of the IP phone
	Headset Wired Port	RJ-11 jack on the bottom of the handset

	Headset	To mount mouthpiece and earpiece on the single handle.
2	LCD Screen	The LCD screen is for displaying your settings, such as phone number, line status and so on.
3	Multi-Functional Key	These keys can be used as speed dial, BLF, shortcut key, pick up and call park.
4	Line Keys	In standby: These keys are used as line keys; you can press the line button to select the corresponding line, and then user can make call or do other functions. The LEDs beneath the keys are used to display the status of each extension.
5	Soft Keys	These keys are used as soft keys. These can be used for item selection or control on the LCD screen. The Soft key function depends on their corresponding content displayed on the LCD at that time.
6	Numeric Keypad	Enters numeric digits for initiating a call or for entering configuration information.
7	Other Function and Numeric keys	Include MENU, UP, DOWN, REDIAL, and Numeric Keys

Rear View

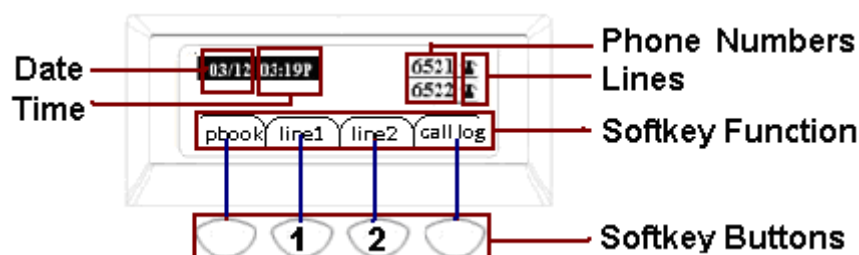


Keypad Descriptions



1	Headset	Headset console, connect to headset
2	DND Switch	The switch is used to turn on or turn off DND. Below the character "DND" is a dot. When the switch is slided to the "dot" area, DND is on; otherwise DND is off. Taking the left picture for example, DND is on.
3	DC 5V,1A	Power port
4	PC	Connects to a PC.
5	Internet	Connects to the Ethernet switch, router or Internet.

Phone Screen Features

This is what your main phone screen might look like with an active call.



Graphic Icon Descriptions

1	Date	To display the current date. Date format is mm/dd
2	Time	To display the current time. Time format is mm:ss (A or P)
3	Phone Numbers	To display the phone number of lines.
4	Lines	To display the status of lines. The icon  means unregistered. The icon  mean registered.
5	Softkey Function	To display the current softkey function.
6	Softkey Buttons	User can use the softkey button to highlight the item displayed on the LCD screen and then configure. One button directs to one softkey function, the blue line in the left

		picture displays the corresponding relationship.
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Chapter 2

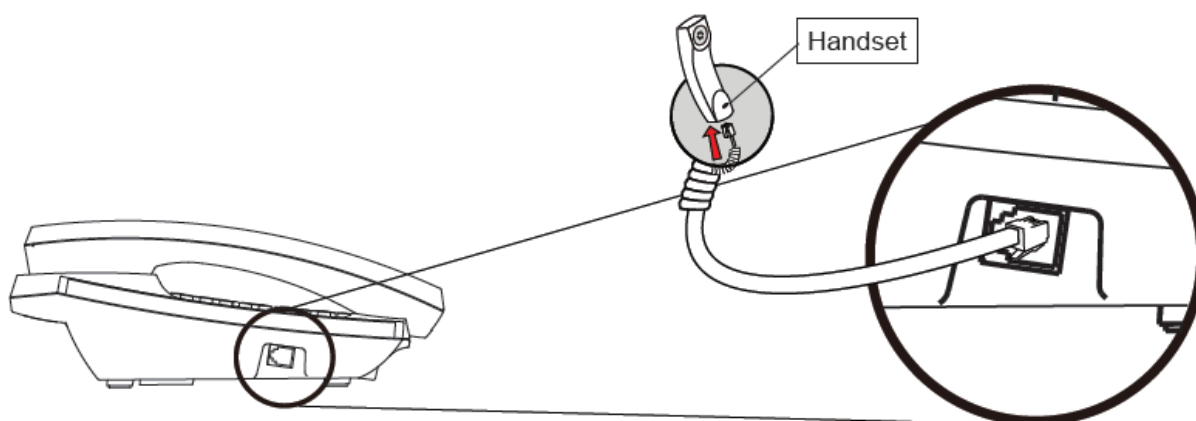
Preparations & Installation

Physical Installation

VIP-256T: Enterprise SIP IP Phone (2 x RJ-45, 1 x Internet interface)

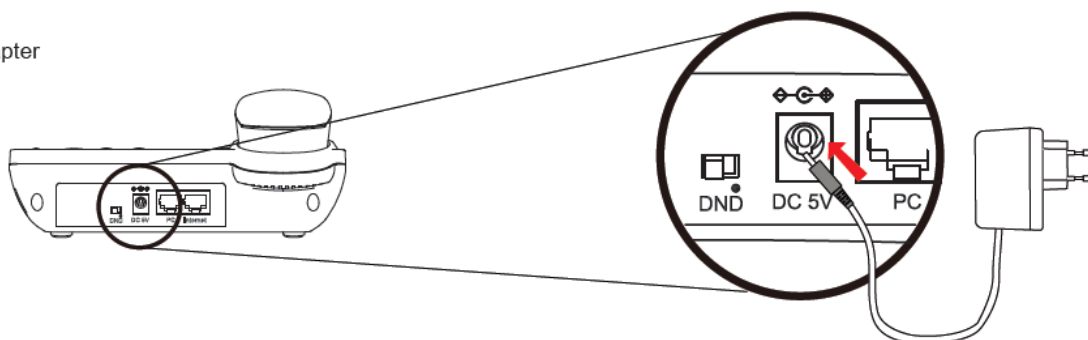
Step 1. Handset Connection

Plug one end of the handset cord into the handset and the other end into the phone body.



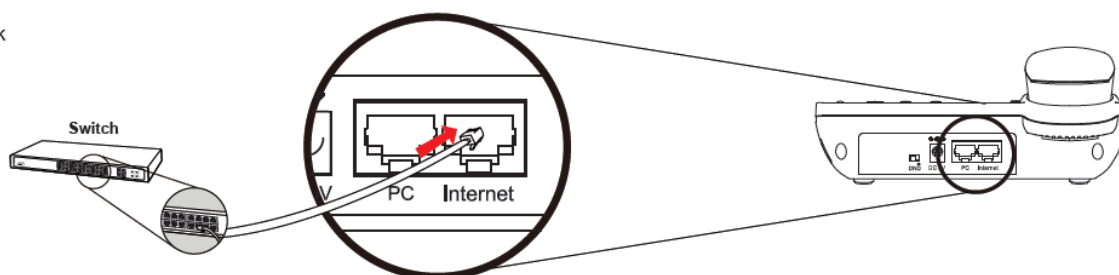
Step 2. Connecting Power Adapter and Network Power Adapter

Power adapter



Network

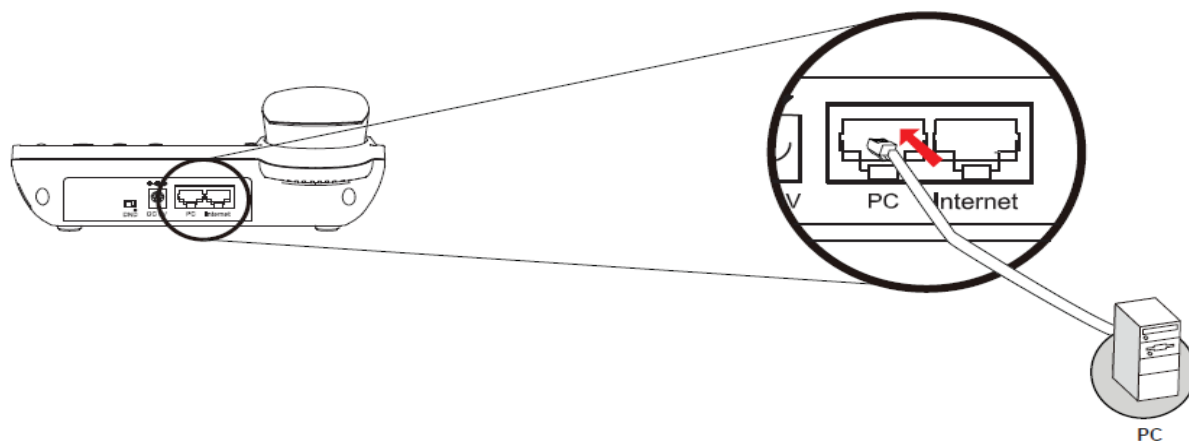
Network



NOTE: Use only the power adapter shipped with the unit to ensure correct functionality

Step 3. Computer Network Setup

Set your computer's IP address to 192.168.0.x, where x is a number between 2 to 254 (except 1 where is being used for the IP Phone by default). If you don't know how to do this, please ask your network administrator.



Step 4. Login Prompt

Use web browser (Internet Explorer 6.0 or above) to connect to 192.168.0.1 (type this address in the address bar of web browser).

You'll be prompted to input user name and password: **admin** and 123

Administration Interface

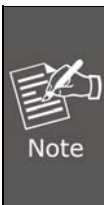
The IP Phone provides GUI (Web based, Graphical User Interface) for machine management and administration. Key pad administration also available for simple configuration.

Web Configuration Access

To start IP Phone web configuration, you must have one of these web browsers installed on computer for management

- Microsoft Internet Explorer 6.0.0 or higher with Java support

Default IP address of IP Phone is **192.168.0.1**. You may now open your web browser, and insert ***http://192.168.0.1*** in the address bar of your web browser to logon IP Phone web configuration page. IP Phone will prompt for logon username and password, please enter: ***admin*** and ***123*** to continue machine administration.



In order to connect machine for administration, please locate your PC in the same network segment (192.168.0.x) of IP Phone. If you're not familiar with TCP/IP, please refer to related chapter on user's manual CD or consult your network administrator for proper network configurations.

Chapter 3

Network Service Configurations

3

Configuring and Monitoring your IP Phone from Web Browser

The IP Phone integrates a web-based graphical user interface that can cover most configurations and machine status monitoring. Via standard, web browser, you can configure and check machine status from anywhere around the world.

Manipulation of IP Phone via Web Browser

Log on IP Phone via web browser

After TCP/IP configurations on your PC, you may now open your web browser, and input <http://192.168.0.1> to logon IP Phone web configuration page.

IP Phone will prompt for logon username and password: **admin** and **123**.



When users login the web page, users can see the IP Phone system information like firmware version, company, etc on this main page.

Chapter 4

VoIP IP Phone Status

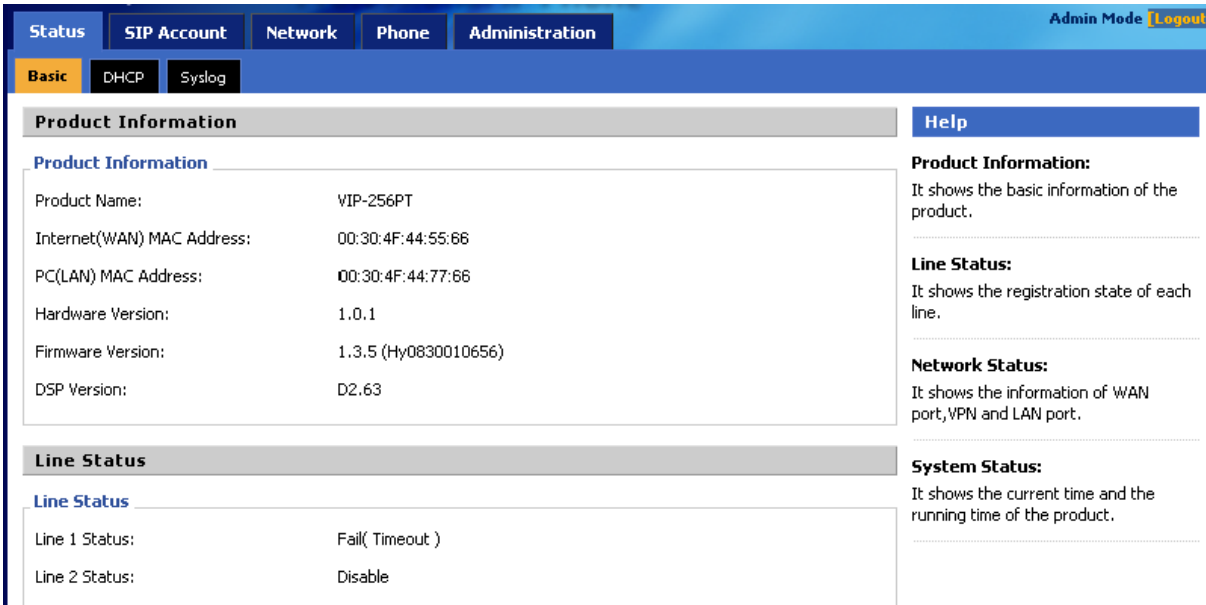
Status

You can check the basic phone status to find out more information about the phone.

They include three parts: Basic, DHCP and Syslog.

Basic

Included on this page are Product Information, Line Status, Network Status, and System Status.



The screenshot shows the web interface for the VoIP IP Phone Status. The top navigation bar includes tabs for Status, SIP Account, Network, Phone, and Administration. The 'Status' tab is active, and within it, the 'Basic' sub-tab is selected. The main content area is divided into two sections: 'Product Information' and 'Line Status'. The 'Product Information' section lists details such as Product Name (VIP-256PT), Internet(WAN) MAC Address (00:30:4F:44:55:66), PC(LAN) MAC Address (00:30:4F:44:77:66), Hardware Version (1.0.1), Firmware Version (1.3.5 (Hy0830010656)), and DSP Version (D2.63). The 'Line Status' section shows the status of two lines: Line 1 Status (Fail(Timeout)) and Line 2 Status (Disable). On the right side, there is a 'Help' section with descriptions for Product Information, Line Status, Network Status, and System Status. The top right corner of the interface shows 'Admin Mode' and a 'Logout' button.

Product Information	
Product Name:	VIP-256PT
Internet(WAN) MAC Address:	00:30:4F:44:55:66
PC(LAN) MAC Address:	00:30:4F:44:77:66
Hardware Version:	1.0.1
Firmware Version:	1.3.5 (Hy0830010656)
DSP Version:	D2.63

Line Status	
Line 1 Status:	Fail(Timeout)
Line 2 Status:	Disable

Help

Product Information:
It shows the basic information of the product.

Line Status:
It shows the registration state of each line.

Network Status:
It shows the information of WAN port, VPN and LAN port.

System Status:
It shows the current time and the running time of the product.

Network Status

Internet Port Status

Connection Status: Connected
 Connection Type: Static IP
 IP Address: 10.1.1.200
 Subnet Mask: 255.255.255.0
 Default Gateway: 10.1.1.254
 Primary DNS: 168.95.1.1
 Secondary DNS: 168.95.1.2

VPN Status

VPN Type: Disable
 Virtual IP Address: 0.0.0.0

PC Port Status

Connection Status: Failed
 Connection Type: Bridge
 IP Address: 192.168.252.1
 Subnet Mask: 255.255.255.0

System Status

System Status

Current Time: Aug 15 12:26:40 2011
 Elapsed Time: 5 D/1 H/30 M

Refresh

Items	Descriptions
Product Information	It shows the basic information of the product.
Line Status	It shows the registration state of each line.
Network Status	It shows the information of Internet port, VPN and PC port.
System Status	It shows the current time and the running time of the product.
Refresh	Click Refresh button to refresh status of phone.

DHCP

This page displays the status about DHCP server enable/disable, start IP address, end IP address and client lease time. Click **Refresh** button to refresh status of DHCP server.

Status	SIP Account	Network	Phone	Administration
Basic	DHCP	Syslog		

Dynamic Host Configuration Protocol

DHCP Status

DHCP Server:	Enable
Start IP Address:	192.168.1.2
End IP Address:	192.168.1.254
Client Lease Time:	48 Hr.

Item	Descriptions
DHCP Status	It shows the information of the DHCP Server.

Syslog

It shows all the log information of system.

Status	SIP Account	Network	Phone	Administration
--------	-------------	---------	-------	----------------

Basic	DHCP	Syslog
-------	------	--------

Syslog

<01/01 00:00:04>***system booting***

<01/01 00:00:07>ip conflict

<01/01 00:00:08>Start Register Client ...

<01/01 00:00:04>***system booting***

<01/01 00:00:10>Start Register Client ...

<01/01 00:00:04>***system booting***

<01/01 00:00:07>ip conflict

<01/01 00:00:08>Start Register Client ...

<02/09 16:48:44>ip conflict

<02/09 16:49:14>ip conflict

<02/09 16:49:44>ip conflict

<02/09 16:50:14>ip conflict

<02/09 16:50:44>ip conflict

<02/09 16:51:15>ip conflict

<01/01 00:00:04>***system booting***

<01/01 00:00:09>Start Register Client ...

<01/01 00:00:04>***system booting***

Chapter 5

SIP Account Setting

SIP Account

SIP is a request-response protocol, dealing with requests from clients and responses from servers. Participants are identified by SIP URLs. Requests can be sent through any transport protocol. SIP establishes call parameters at either end of the communication, and handles call transfer and termination.

SIP Setting

Set your SIP server in the following interfaces. These parameters are related to registration and call.

Status

SIP Account

Network

Phone

Administration

SIP Settings

Line 1

Line 2

SIP Parameters

SIP Parameters

SIP T1: 500 MS

SIP Reg User Agent Name:

Mark All AVT Packets: Enable

SRTP: Disable

Max Forward: 70

Max Auth: 2

RFC 2543 Call Hold: Enable

SRTP Prefer Encrypto: AES_CM

NAT Traversal

NAT Traversal

NAT Traversal: Disable

NAT Refresh Interval(sec): 60

STUN Server Address: stun.fwdnet.net

STUN Server Port: 3478

Items	Descriptions
SIP T1	RFC 3261 T1 value (RTT estimate), which can range from 0 to 64 second. Defaults to .5 seconds
Max forward	SIP Max Forward value, which can range from 1 to 255. Defaults to 70.
SIP Reg User Agent	User-Agent name to be used in a REGISTER request.

Name	If this is not specified, the <SIP User Agent Name> is also used for the REGISTER request. Defaults to blank
Max Auth	Maximum number of times (from 0 to 255) a request may be challenged. Default is 2.
Make ALL AVT Package	For second dial tone, enable this item package by making the position to 1, but 0 for disabling this item.
RFC 254.3 Call Hold	If set to yes, unit will include c=0.0.0.0 syntax in SDP when sending a SIP re-INVITE to the peer to hold the call. If set to no, unit will not include the c=0.0.0.0 syntax in the SDP. The unit will always include a=sendonly syntax in the SDP in either case. Defaults to yes
SRTP	Enable/Disable SRTP(Secure Real-time Transport Protocol)
SRTP Prefer Encrypto	SRTP encryption type.

NAT Setting

Set your NAT Traversal parameters in the following interface. It is helpful for the device behind NAT.

NAT Traversal

NAT Traversal: Disable

NAT Refresh Interval(sec): 60

STUN Server IP: stun.fwdnet.net

Port: 3478

Save Settings
Cancel Changes
Reboot

Item	Descriptions
NAT Traversal	Enable/Disable NAT. VIP-256T supports STUN traversal. Choose "STUN" in the "NAT Traversal Mode" if you want traverse NAT/Firewall.
STUN Server IP	STUN server IP address, default is stun.fwdnet.net
NAT Refresh Interval (sec)	the interval to refresh
Port	STUN port

Line Settings

On this webpage, users can configure the information about SIP account1, including the following 4 parts: Basic, Audio Configuration, User and Advanced, user can program all the SIP parameters. For VIP-256T, it can support 2 lines registered.

Basic

Set the basic information such as Phone Number, Account, Password, SIP Proxy and so on provided by your VOIP Service Provider.

Basic

Basic Setup

Line Enable:	<input type="button" value="Enable"/>	Peer To Peer:	<input type="button" value="Disable"/>
Proxy DNS Type:	<input type="button" value="A Type"/>	VPN:	<input type="button" value="Disable"/>

Proxy and Registration

Domain Name:	<input type="text"/>	SIP Port:	<input type="text" value="5060"/>
SIP Server:	<input type="text" value="192.168.100.100"/>	Outbound Port:	<input type="text" value="5060"/>
Outbound Proxy:	<input type="text"/>		

Subscriber Information

Display Name:	<input type="text" value="6588"/>	Phone Number:	<input type="text" value="6588"/>
Account:	<input type="text" value="6588"/>	Password:	<input type="password" value="••••"/>

Items	Descriptions
Line Enable	Enable/Disable SIP Line
Peer to Peer	Enable/Disable PEER to PEER If enable, SIP line will not send register request to SIP server; In System Status, SIP line Status is Registered; SIP-1 can make call out, but others can not call SIP line.
Proxy DNS Type	Choose DNS type from A Type and DNS SRV.
Use VPN	Enable/Disable VPN
Domain Name	The domain of SIP Server
SIP Server	The IP address of SIP Server
SIP Port	The port which SIP Server supports for VOIP service, default is 5060
Outbound Proxy	Outbound Proxy IP or domain name
Outbound Port	Outbound Proxy's Service port
Display Name	The number will display in callee
Phone Number	Number of telephone provided by SIP Proxy

Account	SIP account provided by SIP Proxy
Password	SIP password provided by SIP Proxy

Audio Configuration

Select the audio Codec you want to use.

Audio Configuration

Codec Setup

Audio Codec Type 1:	<input type="text" value="G.711U"/>	Audio Codec Type 2:	<input type="text" value="G.711A"/>
Audio Codec Type 3:	<input type="text" value="G.729"/>	Audio Codec Type 4:	<input type="text" value="G.722"/>
Audio Codec Type 5:	<input type="text" value="G.723"/>	G.723 Coding Speed:	<input type="text" value="5.3k bps"/>
Packet Cycle(ms):	<input type="text" value="20ms"/>	Echo Cancel:	<input type="text" value="Enable"/>
Silence Supp Enable:	<input type="text" value="Disable"/>		

Items	Descriptions
Audio Codec Type1	Choose the audio codec type from G.711U, G.711A, G.722, G.729, G.723
Audio Codec Type2	Choose the audio codec type from G.711U, G.711A, G.722, G.729, G.723
Audio Codec Type3	Choose the audio codec type from G.711U, G.711A, G.722, G.729, G.723
Audio Codec Type4	Choose the audio codec type from G.711U, G.711A, G.722, G.729, G.723
Audio Codec Type5	Choose the audio codec type from G.711U, G.711A, G.722, G.729, G.723
G.723 Coding Speed	Choose the speed of G.723 from 5.3kbps and 6.3kbps
Packet Cycle	The RTP packet cycle time

Supplementary Services Subscription

Call Waiting - This call feature allows your phone to accept other incoming calls during the conversation.

Supplementary Service Subscription

Supplementary Services

Call Waiting:	Enable ▼	Delayed Hot Line:	
Dial Prefix:		Voice Mailbox Numbers:	
MWI Enable:	Disable ▼		

Items	Descriptions
Call Waiting	Enable / Disable Call Waiting.
Call Pickup	Enable / Disable Call Pickup.
Delayed Hot Line	Fill in the hotline number. Pick up handset or press speaker/headset button, VIP-256T will dial out the hotline number automatically. Ex: xxxT4 will delay 4 seconds, then transfer to xxx (set to T0 will not delay.)
MWI Enable	Enable / Disable MWI (message waiting indicate).
Voice Mailbox Numbers	Fill in the voice mailbox phone number

Advanced Setup

IP phone makes calls based on SIP accounts. IP phone can support 4 independent SIP accounts, and each account can be configured to different SIP servers.

Advanced

Advanced Setup

Domain Name Type:	Disable ▼	Carry Port Information:	Disable ▼
Signal Port:	5060	DTMF Type:	RFC2833 ▼
RFC2833 Payload(>=96):	101	Register Refresh Interval(sec):	3600
RTP Port:	0 (=0 auto select)	Cancel Message Enable:	Disable ▼
Prack Enable:	Disable ▼	SIP Ping Enable:	Disable ▼
Keep-alive Interval(10-60s):	15		

Items	Descriptions
Domain name Mode	If or not use domain name in the SIP URI
Carry Port Information	If or not carry Port information in the SIP URI.
Signal Port	The default of the local port of SIP protocol is 5060
DTMF Type	Choose the DTMF type from IN_band, RFC2833 and SIP INFO.
RFC2833 Payload (>=96)	User can use the default setting
Register Refresh Interval	The interval between two normal Register messages. You can use the default setting.
RTP Port	Set the port to send RTP. IP Phone will select one idle port for RTP if you set "0", otherwise use the value user set.
Cancel Message Enable	When you set enable, an unregistered message will be sent before registration, while you set disable, unregistered message will not be sent before registration. You should set the option for different Proxy.
Prack Enable	Enable / Disable prack.
SIP Ping Enable	If this option enable, IP Phone will send SIP-PING to Server periodically instead of sending hello packet. The send interval is Keep-alive interval.
Keep-Alive Interval (10-60s)	The interval that IP Phone will send an empty packet to Proxy.

Chapter 6

Network Setting

Basic

In this item you can program all the Network parameters.

Status	SIP Account	Network	Phone	Administration
Basic	MAC Address Clone	VPN	DMZ	QoS

Internet Port (WAN)

Internet Port (WAN)

Internet Connection Type: Automatic Configuration - DHCP

DNS Type: Manual

Primary DNS: 168 . 95 . 1 . 1

Second DNS: 168 . 95 . 1 . 2

PC Port(LAN)

PC Port(LAN)

PC Port Connection Type: NAT

Local IP Address: 192 . 168 . 1 . 1

Subnet Mask: 255.255.255.0

Network Address Server Settings (DHCP)

Local DHCP Server: Enable

Start IP Address: 192 . 168 . 1 . 2

Number of Address: 253

Client Lease Time: 48 Hr(0 means one day).

Primary DNS: 219 . 141 . 136 . 10

Second DNS: 219 . 141 . 140 . 10

Internet Port (WAN)

Internet Port WAN (Static IP)

Internet Port (WAN)

Internet Connection Type: Static IP

IP Address: 192 . 168 . 20 . 104

Subnet Mask: 255 . 255 . 255 . 0

Default Gateway: 192 . 168 . 20 . 1

Primary DNS: 202 . 96 . 134 . 33

Second DNS: 202 . 96 . 128 . 86

Items	Descriptions
Internet Connection Type	Choose Static IP.
IP Address	The IP address of Internet port
Subnet Mask	The subnet mask of Internet port.
Default Gateway	The default gateway of Internet port.
Primary DNS	The primary DNS of Internet port.
Second DNS	The second DNS of Internet port.
Internet Connection Type	Choose Static IP.

Internet Port WAN (DHCP)

Internet Port (WAN)

Internet Connection Type: Automatic Configuration - DHCP

DNS Type: Manual

Primary DNS: 202 . 96 . 134 . 33

Second DNS: 202 . 96 . 128 . 86

Items	Descriptions
Internet Connection Type	Choose Automatic Configuration-DHCP.
DNS type	Choose DNS type from Manual to Automatic 1. In Manual: user should set the Primary DNS and Second DNS

	manually. 2. In Automatic: IP Phone will get the Primary DNS and Second DNS from DHCP Server automatically.
--	--

Internet Port WAN (PPPoE)

Internet Port (WAN)

Internet Port (WAN)

Internet Connection Type: PPPoE

PPPoE Account:

PPPoE Password:

MTU:

MRU:

PPPoE Auto Dial: Enable

DNS Type: Manual

Primary DNS: . . .

Second DNS: . . .

Item	Descriptions
Internet Connection Type	Choose PPPoE.
PPPoE Account	Fill in the PPPoE account obtained from Internet Service Provider
PPPoE Password	Fill in the PPPoE account obtained from Internet Service Provider
PPPoE Auto-Dial	Enable / Disable PPPoE Auto-Dial.
DNS Type	Choose DNS type from Manual to Automatic 1. In Manual: user should set the Primary DNS and Second DNS manually. 2. In Automatic: IP Phone will get the Primary DNS and Second DNS from DHCP Server automatically.
Primary DNS	The primary DNS of Internet port.
Second DNS	The second DNS of Internet port.

PC Port (LAN)

Support three modes : disable, NAT and Bridge

PC Port(LAN)

PC Port(LAN)

PC Port Connection Type

NAT

Local IP Address:

192 . 168 . 252 . 1

Subnet Mask:

255.255.255.0

Items	Descriptions
PC Port Connection Type	Choose the PC port connection type from disable, NAT and Bridge. <i>NAT</i> - The product is the same as a router. <i>Bridge</i> - The LAN port is same as the WAN port <i>Disable</i> - PC port switch to NAT mode, but Internet port and PC port can't communication to each other.(The device behind the PC port still can connect to each other)
Local IP Address	Set the IP address of PC port. Efficient when user choose NAT.
Subnet Mask	Set the subnet mask of PC port. Efficient when user choose NAT.

Network Address Server Settings (DHCP)

Support three modes: disable, NAT and Bridge

DHCP Server - It will assign the IP Addressed set here to devices that connect to the LAN port.

Number of Address - You may limit the number of addresses your router hands out.

Network Address Server Settings (DHCP)

Local DHCP Server	Enable <input type="button" value="v"/>
Start IP Address	<input type="text" value="192"/> . <input type="text" value="168"/> . <input type="text" value="252"/> . <input type="text" value="2"/>
Number of Address	<input type="text" value="253"/>
Client Lease Time	<input type="text" value="48"/> Hr (0 means one day).
Primary DNS	<input type="text" value="219"/> . <input type="text" value="141"/> . <input type="text" value="136"/> . <input type="text" value="10"/>
Second DNS	<input type="text" value="219"/> . <input type="text" value="141"/> . <input type="text" value="140"/> . <input type="text" value="10"/>

Items	Descriptions
Local DHCP Server	Enable / Disable DHCP Server. If PC port is not in NAT mode, user can not enable DHCP server.
Start IP Address	The starting IP address which IP phone will attribute to clients. Note: The Network Sect of DHCP Server Start Address should be the same as VIP-256PT's PC port. Generally speaking, you can use the default setting.
Number of Address	Number of IP address will distribute to clients.
Client Lease Time	The interval of DHCP will send request to continue in period of validity. Unit is hour.
Primary DNS	Primary DNS that DHCP Server will distribute. You can use the default setting.
Secondary DNS	Secondary DNS that DHCP Server will distribute. You can use the default setting.
Local DHCP Server	Enable / Disable DHCP Server. If PC port is not in NAT mode, user cannot enable DHCP server.

MAC Address Clone

MAC Address Clone: Some ISPs will require you to register your MAC address. If you do not wish to re-register your MAC address, you can have the router clone the MAC address that is registered with your ISP.

MAC Address Clone

MAC Clone

Clone WAN MAC: 00:00:00:00:00:00

[Get Current PC MAC Address](#)

[Save Settings](#)
[Clear](#)
[Reboot](#)

Items	Descriptions
MAC Clone	<p>MAC is the hardware address of network equipment. Sometimes, network providers may bind network account with the network equipment's MAC address. So you may not pass the provider's authentication when you use a new VIP-256T. In this case, you can use MAC Clone to copy your PC's MAC address to VIP-256T's Internet port.</p> <p>MAC is an important parameter for network equipment, so you should make sure that the MAC is right, in order to prevent to make VIP-256T unusable.</p> <p>You can login VIP-256T's Web via PC port if you are incautious about making it wrong. And then cloning the right MAC or resuming the default settings.</p>
MAC Clone Step	<p>Step 1 Press Get Current PC MAC Address button to get the PC's MAC address</p> <p>Step 2 Press Save Settings to save the changes</p> <p>Step 3. Press Clear to cancel MAC address clone.</p> <p>Step 4. Press Reboot to reboot VIP-256T.</p>

VPN

A Virtual Private Network (VPN) is the extension of a private network that encompasses links across shared or public networks like the Internet. In short, by VPN technology, you can send data between two computers across a shared or public network in a manner that emulates the properties of a point-to-point private link.

VPN Settings

Administration

VPN Enable: PPTP

Initial Service IP: 0.0.0.0

Initial Service Port: 80

User Name: d1

Password: d1

Route Strategy: All

Items	Descriptions
VPN Enable	Enable / Disable VPN. And user can choose the VPN mode from PPTP and L2TP.
Initial Service IP	VPN server IP address.
Initial Service Port	VPN server port.
User Name	The user name for authentication.
Password	Password for authentication.
Route Strategy	Choose route mode from All or SIP.

DMZ

Enabling this option will expose the specified host to the Internet. All ports between the DMZ Start Port and the DMZ End Port will be accessible from the Internet.

Status	SIP Account	Network	Phone	Administration
Basic	MAC Address Clone	VPN	DMZ	QoS

Demilitarized Zone (DMZ)

DMZ

Use DMZ Disable ▾

DMZ Host IP Address

DMZ Start Port

DMZ End Port

Items	Descriptions
Use DMZ	Enable / Disable DMZ
DMZ Host IP Address	set the IP address of DMZ host
DMZ Start Port	set the start port of DMZ host
DMZ End Port	set the end port of DMZ host

DMZ Example:

For example, the DMZ computer's IP is "192.168.1.2". "DMZ start port" and "DMZ end port" are 20 and 1023. The DMZ function is that DMZ computer can get the requests from the ports (20 to 1023) of VIP-256T's Internet port.

QoS

Layer 3 QoS: Set the IP TOS value of SIP and RTP Packets.

Layer 2 Qos: Set the value of 802.1Q and 802.1p priority

Status	SIP Account	Network	Phone	Administration
Basic	MAC Address Clone	VPN	DMZ	QoS

QoS Settings

Layer 3 QoS

SIP QoS	<input type="text" value="0"/>
RTP QoS	<input type="text" value="0"/>
Data QoS	<input type="text" value="0"/>

Layer 2 Qos

802.1Q/VLAN ID	<input type="text" value="0"/>
802.1p PRI	<input type="text" value="0"/>

Items	Descriptions
Some ISP supply QoS services. The QoS services can make the best of improving the quality of Voice application. You can get the settings from the ISP if they supply QoS services. Please contact them if you need it.	

Chapter 7

Phone Configurations

Performance

User can configure the value of ring volume, speakerphone volume, handset volume and so on.

Volume

Volume Settings - Adjust the input gain or the volume of handset/speaker/ring

Preference

Volume Settings

Handset Input Gain:	<input type="text" value="5"/>	Speakerphone Input Gain:	<input type="text" value="5"/>
Handset Volume:	<input type="text" value="5"/>	Speaker Volume:	<input type="text" value="5"/>
Ringer Volume:	<input type="text" value="5"/>		

Items	Descriptions
Handset Input Gain	Adjust the handset input gain from 0-7
Handset Volume Gain	Adjust the output gain from 0-7
Speakerphone Input Gain	Adjust the speakerphone input gain from 0-7
Speaker Volume	Adjust the speaker volume from 0-7
Ringer Volume	Adjust the ringer volume from 0-7.

Regional

Modification of the Tone type and tone parameters.

Regional

Tone Type:	USA ▼	
Dial Tone	<input type="text"/>	
Busy Tone	<input type="text"/>	
Off Hook Warning Tone	<input type="text"/>	
Ring Back Tone	<input type="text"/>	
Call Waiting Tone	<input type="text"/>	
Min Jitter Delay(ms):	<input type="text" value="0"/>	Max Jitter Delay(ms): <input type="text" value="80"/>
Ringing Time(Sec):	<input type="text" value="60"/>	

Items	Descriptions
Tone Type	Choose tone type from China, US, Hong Kong and Korea. Besides custom types, the other types are already defined in the system.
Min Jitter Delay (ms)	The min. value of VIP-256T's jitter delay, VIP-256T's jitter is an adaptive jitter mechanism.
Max Jitter Delay (ms)	The max. value of VIP-256T's jitter delay, VIP-256T's jitter is an adaptive jitter mechanism.
Hook-On Tone Delay (sec)	How long VIP-256T will delay to sound hook-on tone when call party end call.
Ringing Time(Sec)	How long VIP-256T will ring
Busy Tone Delay(Sec)	Before the busy tone VIP-256T will send the delay tone (like di,di.), this parameter define how long the delay tone is.

Call Forward

Call Forward - This feature allows you to forward an incoming call to another phone number.

Call Forward

All Forward:	<input type="text"/>	Busy Forward:	<input type="text"/>
No Answer Forward:	<input type="text"/>	No Answer Timeout:	<input type="text" value="20"/>

Items	Description
All Forward	The phone number which will be forwarded to. IP Phone will forward all calls to the phone number immediately when there is an incoming call.
Busy Forward	The phone number which will be forwarded to when line is busy.
No Answer Forward	The phone number which will be forwarded to when there's no answer at your phone.
No Answer Timeout	The seconds to delay forwarding calls, if there is no answer at your phone.

Miscellaneous

Auto Answer - All the incoming calls will be put through automatically.

Miscellaneous

Auto Answer:	<input type="text" value="Disable"/>	Call Immediately Key:	<input type="text" value="#"/>
Dial Time Out:	<input type="text"/>	Handsfree Key Mode:	<input type="text" value="Handsfree"/>
ICMP Ping:	<input type="text" value="Disable"/>		

Items	Description
Auto Answer	Enable / Disable auto answer. If enable, VIP-256T will auto answer all incoming call immediately.
Dial Time Out	How long VIP-256T to sound dial out tone when VIP-256T dialing number.
Call Immediately Key	Choose call immediately key form * or #.
ICMP Ping	Enable / Disable ICMP Ping. If enable this option, VIP-256T will ping the SIP Server every interval time; otherwise, It will send "hello" empty packet to the SIP Server.

Handsfree Key Mode	Choose the handsfree key mode from handsfree and headset.
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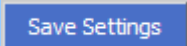

Multi-Functional Key

In here user can program the Multi-Function Key like Speed dial , BLF, Shortcut Key, Call Pick up

Status	SIP Account	Network	Phone	Administration
Preference	Multi-Functional Key	Dial Plan	Phonebook	Call Log

Multi-Functional Key					
Key	Type	Mode	Line	Expansion	Pickup Code
Exp Key 1	Disable	Phonebook	Line 1		
Exp Key 2	Disable	Phonebook	Line 1		
Exp Key 3	Disable	Phonebook	Line 1		
Exp Key 4	Disable	Phonebook	Line 1		
Exp Key 5	Disable	Phonebook	Line 1		
Exp Key 6	Disable	Phonebook	Line 1		
Exp Key 7	Disable	Phonebook	Line 1		
Exp Key 8	Disable	Phonebook	Line 1		
Exp Key 9	Disable	Phonebook	Line 1		
Exp Key 10	Disable	Phonebook	Line 1		

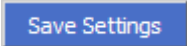

Items	Descriptions
Program Step:	
	Step 1.Choose one current key board to configure from Basic Board, Expansion Board 1, Expansion Board 2, Expansion Board 3, Expansion Board 4, Expansion Board 5 and Expansion Board 6.
	Step 2.Choose one Exp Key from Exp Key 1 to Exp Key 20.
	Step 3.Choose one function type from Speed Dial, BLF, Shortcut Key and Call Pick Up.
	Step 4.Set the other corresponding parameters.
	Step 5.Press Save Settings button to save changes, press Cancel Changes button to cancel changes.
Adding speed dial:	
	Speed Dial: You can configure the key as simplified speed dial key.This key function allows you to easily access the most frequently dialed numbers.

1. Choose one Exp Key to configure
2. Select the speed dial from the drop down list
3. Choose the Line from auto (the first line registered), line1, line2, line3, line 4 and line 5
4. Fill the phone number in Expansion
5. Press  to save changes and you can see the status of corresponding LED is solid green.
6. Press  button to make changes effective.

If set properly, press the corresponding key to make call immediately, and the status of LED is solid red.

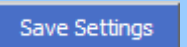
Adding BLF:

BLF: The button can be configured to Busy Line Field function with specified account. This feature must be supported by the sip server.

- 1) Choose one Exp Key to configure
- 2) Select the BLF from the drop down list
- 3) Choose the Line from line1, line2, line3, line 4 and line 5.
- 4) Fill the monitored phone number in Expansion
- 5) Fill the pickup code in Pickup Code if user wants to pickup the call when there is a new call coming in monitored phone.
- 6) Press  to save changes and you can see the status of corresponding LED is solid green.
- 7) Press  button to make changes effective.

Adding shortcut key:

Shortcut Keys: Shortcut Keys are predefined shortcuts to phone and call functions.

1. Choose one Exp Key to configure
2. Select the shortcut key from the drop down list
3. Select the mode from the phonebook, call history, text message, volume+, volume- and login/logout in the drop down list.
4. Press  to save changes and you can see the status of corresponding LED is

solid green.

5. Press **Reboot** button to make changes effective.
6. If set properly, press the corresponding button to access to phonebook, call history, text message, volume+, volume- and login/logout menu directly.

Adding call park:

1. Choose one Exp Key to configure
2. Select the Call Park from the drop down list in type
3. Choose the Line from line1, line2, line3, line 4 and line 5.
4. Fill the pickup extension code in Expansion
5. Press **Save Settings** to save changes and you can see the status of corresponding LED is solid green.
6. Press **Reboot** button to make changes effective.

Dial Plan

Dial Plan

General
Dial Plan Disable

No.	Line	Digit Map	Action	Move Up	Move Down	
Line	Line 1	<div style="border: 1px solid #ccc; height: 20px; width: 150px;"></div>	Deny			

OK
Cancel

Save Settings
Cancel Changes
Reboot

Items	Descriptions
Dial Plan	Enable / Disable dial rule.
Line	Choose the call mode from line1, line2, line3, line4 and line5.
Digit Map	Fill in the sequence used to match input number

	For the syntactic, please refer to the following Dial Plan Syntactic
Action	Choose the dial plan mode from Deny and Dial Out. Deny means VIP-256T will reject the matched number, while Dial Out means VIP-256T allow dial out the matched number.
Move Up	Press it to move up.

Dial Plan

General

Dial Plan

Disable

No.	Line	Digit Map	Action	Move Up	Move Down	
1	Line1	<9:010>2010110	Dial Out	▲	▼	<input type="checkbox"/>
2	Line2	<5,:><:241333>8101	Dial Out	▲	▼	<input type="checkbox"/>
3	Line3	<[4-6]:>22x<:333>	Dial Out	▲	▼	<input type="checkbox"/>
4	Line4	<9,8,:>711	Dial Out	▲	▼	<input type="checkbox"/>
5	Line5	<[2-5],:5>622.	Deny	▲	▼	<input type="checkbox"/>

Line

Line 1

Digit Map

Action

Deny

OK

Cancel

Save Settings

Cancel Changes

Reboot

Items	Descriptions
Adding one dial plan:	
Step 1. Enable Dial Plan	
Step 2. Click Add button, and the configuration table	
Step 3. Fill in the value of parameters.	
Step 4. Press OK button to end configuration.	
Step 5. Press Save Settings button to save changes.	
Editing one dial plan:	
Step 1. Enable Dial plan	
Step 2. Choose one dial plan	

- Step 3. Click Edit button, and the configuration table
- Step 4. Change the value of parameters.
- Step 5. Press OK button to end configuration.
- Step 6. Press Save Settings button to save changes.

Deleting one dial plan:

- Step 1. Enable Dial plan
- Step 2. Choose one dial plan
- Step 3. Click Delete button to delete the dial plan

Dial Plan Syntactic

Items	Descriptions
0 1 2 3 4 5 6 7 8 9 * #	Legal characters
X	Lowercase letter x stands for one legal character
[sequence]	To match one character form sequence. For example: 1. [0-9]: match one digit form 0 to 9 2. [23-5*]: match one character from 2 or 3 or 4 or 5 or *
x.	Match to $x^0, x^1, x^2, x^3, \dots, x^n$ For example: "01.": can match "0", "01", "011", "0111",, "01111..."
<diald: substituted>	Replace diald with substituted. For example : <8:1650>123456 : input is "85551212", output is "16505551212"
x,y	Make outside dial tone after dialing "x", stop until dialing character "y" For example : "9,1xxxxxxxxx":VIP-256T make outside dial tone after inputting "9", stop tone until inputting "1" "9,8,010x": make outside dial tone after inputting "9", stop tone until inputting "0"
T	Set the delayed time. For example: "<9:111>T2": VIP-256T will dial out the matched number "111" after 2 seconds.

Dial Plan

General
Dial Plan Enable

No.	Line	Digit Map	Action	Move Up	Move Down	
1	Line1	<:010>#12<#: %23>2	Dial Out	▲	▼	<input type="checkbox"/>
2	Line2	<5,;:><:241333>8101	Dial Out	▲	▼	<input type="checkbox"/>
3	Line3	<[4-5]:>22xxxx<:333>	Dial Out	▲	▼	<input type="checkbox"/>
4	Line4	<2-3,;5:>622,	Dial Out	▲	▼	<input type="checkbox"/>
5	Line5	777x.8	Deny	▲	▼	<input type="checkbox"/>

Example 1
Example 2

Example 5

Example 3
Example 4

Items	Descriptions
Example 1	If user dials #12#2, VIP-256T will call 010#12%232 immediately.
Example 2	If user dials 5,8101, VIP-256T will call 2413338101 immediately, And VIP-256T will make outside dial tone after inputting “5”, stop tone until inputting “8”.
Example 3	If user dials 422xxxx or 522xxxx, VIP-256T will call 22xxxx333 immediately.
Example 4	If user dials 2,622 or 2,6222 or 2,62222 or 2.622222 or 3.622222 , VIP-256T will call 5622 or 56222 or 562222 or 5622222 or 5622222 immediately. And VIP-256T will make outside dial tone after inputting “2” or “3”, stop tone until inputting “6”.
Example 5	If user dials 777x8 , VIP-256T will reject the phone number.

Phonebook

Phonebook

The list shows all the directory entries. Please click "Save Settings" button to save this list after you edit or add an item.

Name

Number

Items	Descriptions
Name	Input the name
Number	Input the phone number

Phonebook

Index	Name	Number	
1	amm	111	<input type="checkbox"/>
2	bob	112	<input type="checkbox"/>
3	tom	113	<input checked="" type="checkbox"/>
4	alice	114	<input type="checkbox"/>
5	lily	115	<input type="checkbox"/>
6	arice	116	<input type="checkbox"/>
7	jon	117	<input type="checkbox"/>
8	wic	118	<input type="checkbox"/>
9	wali	119	<input type="checkbox"/>
10	luce	120	<input type="checkbox"/>

Items	Descriptions
Adding one phone book: Step 1. Click Add button, and the configuration table Step 2. Fill in the value of parameters. Step 3. Press OK button to end configuration.	

Step 4. Press Save Settings button to save changes.

Editing one phone book:

Step 1. Choose one phone book

Step 2. Click Edit button, and the configuration table

Step 3. Change the value of parameters.

Step 4. Press OK button to end configuration.

Step 5. Press Save Settings button to save changes.

Deleting one phone book:

Step 1. Choose one phone book

Step 2. Click Delete button to delete the phone book

Move one phone book to Black list:

Step 1. Choose one phone book

Step 2. Click Move to blacklist button to delete the phone book

Black List

Calls from this list cannot get through.

Name

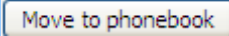
Number

Items	Descriptions
Name	Input the name
Number	Input the phone number

Black List			
Index	Name	Number	
1	k	122	<input type="checkbox"/>
2	w	123	<input checked="" type="checkbox"/>
3	q	124	<input type="checkbox"/>
4	r	125	<input type="checkbox"/>

Name

Number

Items	Descriptions
	Adding one Black List: Step 1. Click Add button, then the configuration table. Step 2. Fill in the value of parameters. Step 3. Press OK button to end configuration. Step 4. Press Save Settings button to save changes.
	Editing one Black List: Step 1. Choose one black list Step 2. Click Edit button, and the configuration table Step 3. Change the value of parameters. Step 4. Press OK button to end configuration. Step 5. Press Save Settings button to save changes.
	Deleting one Black List: Step 1. Choose one black list Step 2. Click Delete button to delete the black list
	Moving one Black List to phonebook: Step 1. Choose one black list Step 2. Click  button to move the black list to the phonebook

Call Log

To view the call log information such as redial list (incoming call), answered call and missed call

Status

SIP Account

Network

Phone

Administration

Preference

Multi-Functional Key

Dial Plan

Phonebook

Call Log

Redial List

Index	Name	Number	Start Time	Duration	<input type="checkbox"/>
1	1000	1000	09/02 17:56	00:00:33	<input type="checkbox"/>
2	1000	1000	09/02 17:54	00:00:00	<input type="checkbox"/>
3	1001	1001	09/02 17:54	00:00:11	<input type="checkbox"/>

Answered Calls

Index	Name	Number	Start Time	Duration	<input type="checkbox"/>
1	1000	1000	09/02 17:54	00:00:22	<input type="checkbox"/>

Chapter 8

VoIP IP Phone Administration

Management

On this page, configuration of the value of Time/Date, password, web access, and system log and so on is made.

Time/Date

Items	Description
NTP Server	Fill in the NTP server IP address or Domain name
Time Zone	Choose the time zone
Manual Time	Adjust time by manual
Alarm Enable	If or not enable alarm
Alarm Time	Set alarm time
Daylight Saving Time	If or not enable daylight saving time.
Offset	Offset time, “-60” means advancing 60miniter, “60” means delaying 60minite
Start Month	Choose starting month
Start Day of Week	Choose starting day
Start Day of Week Last in Month	Choose starting week
Start Hour of Day	Choose starting hour
Stop Month	Choose stopping month
Stop Day of Week	Choose stopping day
Stop Day of Week Last in Month	Choose stopping week
Stop Hour of Day	Choose stopping the function hour

Time/Date

Time/Date

NTP Server:

Time Zone:

Manual Time: : :

Alarm Enable:

Alarm Time: : :

Daylight Saving Time

Offset Min.

Start Month

Start Day of Week

Start Day of Week Last in Month

Start Hour of Day

Stop Month

Stop Day of Week

Stop Day of Week Last in Month

Stop Hour of Day

Items	Descriptions
Alarm Setting: Step 1. Enable alarm Step 2. Set alarm time Step 3. Press Save Settings button to save changes and then press Reboot button to active changes	

Alarm Enable:

Alarm Time: : :

Items	Descriptions
Daylight Saving Time: Step 1. Enable Daylight Saving Time. Step 2. Set value of offset, Step 3: Set starting Month/Week/Day/Hour in Start Month/Start Day of Week Last in Month/Start Day of Week/Start Hour of Day, analogously set stopping Month/Week/Day/Hour in Stop Month/Stop Day of Week Last in Month/Stop Day of Week/Stop Hour of Day. Step 5. Press Saving Settings button to save and press Reboot button to active changes.	

Daylight Saving Time	Enable ▾
Offset	60 Min.
Start Month	March ▾
Start Day of Week	Sunday ▾
Start Day of Week Last in Month	Last in Month ▾
Start Hour of Day	2
Stop Month	October ▾
Stop Day of Week	Sunday ▾
Stop Day of Week Last in Month	Last in Month ▾
Stop Hour of Day	3

Password Reset

Items	Descriptions
User Type	Choose the user type from admin and user.
Original Password	Input original password
New Password	Input the new password
Password Confirm	Input the new password again

Password Reset

Password Reset

User Type	admin ▾
Original Password:	<input type="text"/>
New Password:	<input type="text"/>
Confirm Password:	<input type="text"/>

Items	Descriptions
Change the password: Step 1. Choose the admin from the drop-down list. Step 2. Input original password, default setting is null. Step 3. Input a new password twice time in New Password and Confirm	

Web Access

Items	Description
WAN Interface Login	If or not enable user login WEB via Internet port. If enable, user can access Web to administration.
Web Login Port	Set the port which used to login WEB via Internet port and PC port, Default is 8080, that is why URL should have 8080.
Web Idle Timeout	Set the web idle timeout time. The web page can be logged out after Web Idle Timeout without any operation.

Web Access:

Web Access:

WAN Interface Login:

 Web Login Port:

 Web Idle Timeout: Min.

System Log Setting

Items	Descriptions
SysLog Server	Set the SysLog Server IP address or domain name for VIP-256PE.
Log Level	Choose log level from None/Error/Warn/INFO/Debug. The priority changes from left to right, left is the lowest, right is the highest; the higher priority, the more information in syslog.

System Log Setting

Syslog Server:

 Log Level:

Local and remote Syslog

In local:

- Step 1. Set syslog server null and choose one kind of Log Level.
- Step 2. Press Saving Settings button to save and press Reboot button to active changes.
- Step 3. User can view syslog in Status/Syslog webpage.

In remote:

- Step 1. Fill in syslog server IP address or domain name
- Step 2. Choose one kind of Log Level.
- Step 3. Press Saving Settings button to save and press Reboot button to active changes.
- Step 4. User can view syslog in syslog server, and you can also view the syslog in Status/Syslog webpage.

System Log Setting

System Log Setting

Syslog Server:

Log Level: DEBUG 

Status	SIP Account	Network	Phone	Administration
Basic	DHCP	Syslog		

Syslog

```

<01/01 00:00:04>***system booting***
<01/01 00:00:07>ip conflict
<01/01 00:00:08>Start Register Client ...
<01/01 00:00:04>***system booting***
<01/01 00:00:10>Start Register Client ...
<01/01 00:00:04>***system booting***
<01/01 00:00:07>ip conflict
<01/01 00:00:08>Start Register Client ...
<02/09 16:48:44>ip conflict
<02/09 16:49:14>ip conflict
<02/09 16:49:44>ip conflict
<02/09 16:50:14>ip conflict
<02/09 16:50:44>ip conflict
<02/09 16:51:15>ip conflict
<01/01 00:00:04>***system booting***
<01/01 00:00:09>Start Register Client ...
<01/01 00:00:04>***system booting***

```


Factory Defaults

Items	Description
Factory Default	Press Factory Default button to set VIP-256PE to default.

Factory Defaults:

Reset to Factory Default:

Factory Default

Save Settings

Cancel Changes

Reboot

Updating Firmware

Click on the *Browse* button to select the firmware file to be uploaded to the router.

Firmware Management

Firmware Upgrade

Upgrade Types:

Upgrade Software

Local Upgrade:

Browse...

Security

CA Certificate - The issuer of the certificate.

Client Certificate - user's certificate issued by CA.

Private Key - user's private key file.

Items	Description
TR069 CA Certificate	The CA certificate file of TR069
TR069 Client Certificate	The Client Certificate file of TR069
TR069 Private Key	The Private Key file of TR069
Provision CA Certificate	The CA certificate file of provision
Provision Client Certificate	The Client Certificate file of provision
Provision Private Key	The Private Key file of provision

Certificate Update

Update Type:

Local Upload:

Upload TR069 and Provision

User can upload cert files for TR069 and Provision as follows:

- Step 1. Choose one File Type from
- Step 2. Press to browser file.
- Step 3. Press to start upgrading.

Next is the webpage where all files have well been uploaded.

Status	SIP Account	Network	Phone	Administration
Management	Firmware Upgrade	Security	Provision	SNMP
TR069				

Certificate Management

TR069			
	Issued To	Issued By	Expiration
CA Certificat	none	none	none
Client Certificat	none	none	none
Private Key	none		

Provision

	Issued To	Issued By	Expiration
CA Certificat	none	none	none
Client Certificat	none	none	none
Private Key	none		

Certificate Update

Update Type:

Local Upload:

Provision

Provision allows a device automatically resync to a specific configuration file on a TFTP server or a web server which use HTTP or HTTPS.

- 1) Provisioning allow VIP-256T auto-upgrading or auto-configuring
- 2) VIP-256T supports 3 ways to provision: TFTP, HTTP and HTTPS.
 - ◆ Before testing or using TFTP, user should have TFTP server and upgrading file and configuring file.
 - ◆ Before testing or using HTTP, user should have HTTP server and upgrading file and configuring file.
 - ◆ Before testing or using HTTPS, user should have HTTPS server and upgrading file and configuring file and CA Certificate file(should same as https server's) and Client Certificate file and Private key file
- 3) User can uploading CA Certificate file and Client Certificate file and Private Key file on Equipment Manage/Cert Manage page.

Items	Descriptions
Provision Enabled	If or not enable provision
Resync On Reset	If or not enable resync after VIP-256T restart
Resync Random Delay	Set the maximum delay for request the synchronization file
Resync Periodic	Set the periodic time for resync, default is 3600s
Resync Error Retry Delay	If the last resync was failure, VIP-256T will retry resync after the "Resync Error Retry Delay" time, default is 3600s
Forced Resync Delay	If it's time to resync, but VIP-256T is busy now, in this case, VIP-256T will wait for a period time, the longest is "Forced Resync Delay", default is 14400s, when the time over, VIP-256T will forced to resync
Resync After Upgrade Attempt	If or not enable firmware upgrade after resync, "yes" is enable
Profile Rule	URL of profile provision file
Phone Num1 for Config	The first phone number which used to reboot VIP-256T in remote.
Phone Num2 for Config	The second phone number which used to reboot VIP-256T in remote.
Auto-upgrade Enabled	If or not enable firmware upgrade.
Auto-upgrade Error Retry Delay	Set the time to retry upgrade, effective when the last upgrade was failure
Upgrade Rule	URL of upgrade file

Status	SIP Account	Network	Phone	Administration
Management	Firmware Upgrade	Security	Provision	SNMP
			TR069	

Provision

Configuration Profile

Provision Enable	<input type="text" value="yes"/>	Resync On Reset	<input type="text" value="yes"/>
Resync Random Delay	<input type="text" value="40"/>	Resync Periodic	<input type="text" value="3600"/>
Resync Error Retry Delay	<input type="text" value="3600"/>	Forced Resync Delay	<input type="text" value="14400"/>
Resync After Upgrade Attempt	<input type="text" value="yes"/>		
Profile Rule	<input type="text"/>		
Private Key Password:	<input type="text" value="whatever"/>		
Phone Num1 for Config	<input type="text"/>		
Phone Num2 for Config	<input type="text"/>		

Firmware Upgrade

Upgrade Enable	<input type="text" value="yes"/>
Upgrade Error Retry Delay	<input type="text" value="3600"/>
Downgrade Rev Limit	<input type="text" value="0"/>
Upgrade Rule	<input type="text"/>

SNMP

Allow the device to be managed by the Manager which is set in the SNMP Manager IP.

Items	Descriptions
SNMP Enable	If or not enable SNMP
Get Community	String, as an express password between management process and the agent process
Set Community	String, as an express password between management process and the agent process
SNMP Manager IP 1-4	The IP address of SNMP Manager



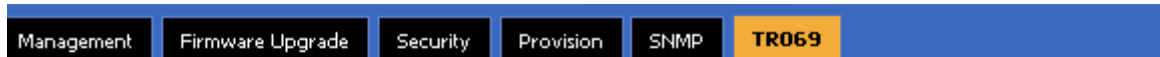
SNMP Configuration

SNMP Configuration

SNMP Service:	Enable <input type="button" value="v"/>
Read Community Name:	<input type="text"/>
Write Community Name:	<input type="text"/>
SNMP Manager IP 1:	<input type="text"/>
SNMP Manager IP 2:	<input type="text"/>
SNMP Manager IP 3:	<input type="text"/>
SNMP Manager IP 4:	<input type="text"/>
SNMP Trap Server IP:	<input type="text"/>

TR 069

Allow the device to be managed by the ACS server which is set in the ACS URL.



TR069 Configuration

ACS

TR069 Enable:	<input type="button" value="Disable"/> ▼
CWMP	<input type="button" value="Enable"/> ▼
ACS URL	<input type="text"/>
User Name	<input type="text"/>
Password	<input type="text"/>
Periodic Inform Enable	<input type="button" value="Enable"/> ▼
Periodic Inform Interval	<input type="text" value="30"/>

Connect Request

User Name	<input type="text"/>
Password	<input type="text"/>
SSL Key	<input type="text"/>

Items	Descriptions
TR069 Enable	If or not enable TR069
CWMP	If or not enable TR069
ACS URL	The URL of TR069 server
User Name	The VIP-256T's user name for connecting to TR069 server
Password	The VIP-256T's password for connecting to TR069 server
Periodic Inform Enable	If or not enable periodic information
Periodic Inform Interval	The interval to send information to TR069 server
User Name	The TR069 server's user name for connecting to VIP-256T
Password	The TR069 server's password for connecting to VIP-256T
SSL Key	Fill in SSL key.

Appendix A -- Frequently Asked Questions

Q1: No Operation after Power On?

A1: Check if the power adapter is properly connected.

Q2: No Dial Tone?

A2: Check if the handset cord is properly connected.

Q3: Cannot Make a Call?

A3: Check the status of your SIP registration status or contact your administrator, supplier, or ITSP for more information or assistance.

Q4: Cannot Receive Any Phone Call?

A4 : Check the status of your SIP registration status, or contact your administrator, supplier, or ITSP for more information or assistance

Q5: No Voice during an Active Call?

A5: Check if the servers support the current audio codec type, or contact your administrator, supplier, or ITSP for more information or assistance.

Q6: Cannot connect to the configuration Website?

A6: Check if the Ethernet cable is properly connected.

Check if the URL is correctly written. The format of URL is: http:// the Internet port IP address

Check if your firewall/NAT settings are correct.

Check if the version of IE is IE8, or use other browser such as Firefox or Mozilla, or contact your administrator, supplier, or ITSP for more information or assistance.

Q7: Forget the Password?

A7: Default password of website and menu is null.

If user changed the password and then forgot it, you cannot access to the configuration website or the menu items which need password.

Solution:

Factory default: press Menu button and choose 16Factory Default, then a notice will appear, choose OK by using the corresponding softkey button.

If you choose factory default, you will return the phone to the original factory settings and will erase ALL current settings, including the directory and call logs.

Q8: How to switch to a different line to dial out?

A8: Before dialing out, press the correspondence line number you want to use, eg.,want to use Line 4

to dial out, must press 2 to switch to line 2, then dial out.



Q9: Why my MSG light would not show any information when I receive the message?

A9: In SIP accounts / Lines / Supplementary service

Please enable the MWI (Message Waiting Indicate) and in Voice Mailbox numbers, please also assign the Voice mail number.

Supplementary Service Subscription

Supplementary Services

Call Waiting:

Dial Prefix:

Hot Line:

MWI Enable:

Voice Mailbox Numbers:

EC Declaration of Conformity

For the following equipment:

*Type of Product : 802.3af PoE SIP IP Phone (2-Line)

*Model Number : VIP-256T / VIP-256PT

* Produced by:

Manufacturer's Name : **Planet Technology Corp.**

Manufacturer's Address: 10F., No.96, Minquan Rd., Xindian Dist., New Taipei City 231,
Taiwan (R.O.C.)

is hereby confirmed to comply with the requirements set out in the Council Directive on the Approximation of the Laws of the Member States relating to Electromagnetic Compatibility Directive (2004/108/EC), For the evaluation regarding the Electromagnetic Compatibility (2004/108/EC), the following standards are applied:

EN 5502	2006+A1:2007
EN 55024	1998+A1:2001+A2:2003
EN 61000-3-2	2006+A2:2009
EN 61000-3-3	2008

Responsible for marking this declaration if the:

☒ Manufacturer ☐ Authorized representative established within the EU

Authorized representative established within the EU (if applicable):

Company Name: Planet Technology Corp.

Company Address: 10F., No.96, Minquan Rd., Xindian Dist., New Taipei City 231, Taiwan
(R.O.C.)

Person responsible for making this declaration

Name, Surname Jonas Yang

Position / Title : Product Manager

Taiwan

Place

30th Mar., 2012

Date



Legal Signature

PLANET TECHNOLOGY CORPORATION